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Foreword

Part 4 of this final draft Interim European Telecommunication Standard (I-ETS) has been produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI), and is now submitted for the Voting phase of the ETSI standards approval procedure.

An ETSI standard may be given I-ETS status either because it is regarded as a provisional solution ahead of a more advanced standard, or because it is immature and requires a "trial period". The life of an I-ETS is limited to three years after which it can be converted into an ETS, have it's life extended for a further two years, be replaced by a new version, or be withdrawn.

This I-ETS is part 4 of a multipart standard covering "Integrated Services Digital Network (ISDN); Videotelephony teleservice", as described below:

Part 1: "Electroacoustic characteristics for handset telephony function when using Pulse Code Modulation encoding".

Part 2: "Audio aspects - Pulse Code Modulation (PCM) A-law loudspeaking and handsfree".

Part 3: "Audio aspects - wideband handset".

Part 4: "Audio aspects - wideband coding and loudspeaking or handsfree function".

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1 Scope

This Interim European Telecommunication Standard (I-ETS) specifies the electroacoustic characteristics for 7 kHz bandwidth loudspeaking and handsfree functions implemented in videotelephony terminals intended for use in the videotelephone teleservice in the Integrated Services Digital Network (ISDN) using Adaptive Differential Pulse Code Modulation (ADPCM) encoding according to CCITT Recommendation G.722 (see annex A). Those terminals will be connected to the basic access of the coincident S and T reference point of the ISDN.

The videotelephony teleservice in the ISDN is described in ETS 300 264 (see annex A).

The requirements of this I-ETS specify those characteristics which deviate from those of an ISDN 7 kHz telephony terminal. Those deviations are due to conditions which are special for videotelephony applications (e.g. delay, measurement position). The corresponding requirements for an ISDN 7 kHz telephony loudspeaking and handsfree terminal can be found in I-ETS 300 245-6 [1].

2 Normative references

This I-ETS incorporates by dated and undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this I-ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to applies.

[1] ETS 300 245-6: "Integrated Services Digital Network (ISDN); Technical characteristics for telephony terminals; Part 6: Wideband (7 kHz) loudspeaking and handsfree telephony".

[2] ETS 300 145 (Second edition): "Integrated Services Digital Network (ISDN); Audiovisual services; Videotelephone systems and terminal equipment operating on one or two 64 kbit/s channels".

[3] I-ETS 300 245-5: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 5: Wideband (7 kHz) handset telephony".

[4] I-ETS 300 245-3: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 3: Pulse Code Modulation (PCM) A-law, loudspeaking and handsfree telephony".

[5] Draft prI-ETS 300 302-3: "Integrated Services Digital Network (ISDN); Videotelephony teleservice; Part 3: Audio aspects - wideband handset".

[6] CCITT Recommendation G.711 (1990): "Pulse code modulation (PCM) of voice frequencies".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this part of the I-ETS, the following definitions apply:

Acoustic Reference Level (ARL): The acoustic level which gives - 10 dBm0 at the digital interface.

hands-free function: For free handling, no handset or any other equipment with transducer is held to the ear of the user. If a handset is implemented then it is placed at a distance from the user. Normally, the handset is not active. The number, the implementation and the use of microphone(s) and loudspeaker(s) are not limited.

Hands-Free Reference Point (HFRP): A point localised on the axis of the artificial mouth, at 50 cm from the lip ring, where the calibration is made, in free field. It corresponds to the measurement point n°11 defined in ITU-T Recommendation P.51 (see annex A).

idle mode: When the terminal is not activated by an input signal (e.g. input signal level below implemented threshold level).

loudspeaking function: The handset is used in the normal position. The incoming signal is simultaneously presented to the user(s) from the loudspeaker(s).

Loudness Rating Guard Ring Position (LRGP): A definition of this position can be found in ITU-T Recommendation P.64 (see annex A).

modes of operation: For the videotelephony teleservice in the ISDN, the modes of operation are listed in subclause 5.3.4 and table 3 of ETS 300 145 [2].

telephony 3,1 kHz teleservice: A teleservice providing speech transmission at an audio bandwidth of 3,1 kHz. The communication is bi-directional, with both directions active during the speech phase. User information provided over a B-channel, signalling is provided over the D-channel (based on ETS 300 111, clause 5).

Terminal Coupling Loss (TCL): The frequency dependent coupling loss between the receiving port and sending port of a terminal due to:

- acoustical coupling at the user interface;
- electrical coupling due to crosstalk in the handset cord or within the electrical circuits;
- seismic coupling through the mechanical parts of the terminal.

NOTE 1: The receiving port and the sending port of a digital voice terminal is a 0 dBr point.

NOTE 2: The coupling at the user interface depends on the conditions of use.

terminal types: For the videotelephony teleservice in the ISDN, terminal types are defined in ETS 300 145 [2].

videotelephony teleservice: A real-time audiovisual teleservice in which speech and moving pictures are interchanged by means of one or two 64 kbit/s circuit mode connections in the ISDN. It is characterised by the transmission of moving pictures simultaneously with speech (based on ETS 300 264, clause 5).

Weighted Terminal Coupling Loss (TCLw): The weighted Terminal Coupling Loss using the weighting of CCITT Recommendation G.122 (see annex A).

3.2 Abbreviations

For the purposes of this Part of the I-ETS, the following abbreviations apply:

ARL	Acoustic Reference Level
HFT	Hands-Free Terminal
HFRP	Hands-Free Reference Point
LRGP	Loudness Rating Guard Ring Position
RLR	Receiving Loudness Rating
S/D	Signal to Distortion
SLR	Sending Loudness Rating
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss

4 Requirements

The technical requirements given in I-ETS 300 245-6 [1] and I-ETS 300 245-5 [3] shall apply with the exception of the following aspects.

4.1 General

4.1.1 Encoding

The normal working mode of the terminal shall be the b2 mode as defined in ETS 300 145 [2], i.e. the speech is transmitted at 56 kbit/s or 48 kbit/s audio coding in one B-channel and video image is transmitted at 76,8 kbit/s or 68,8 kbit/s.

Implementation of 64 kbit/s coding for CCITT Recommendation G.722 (see annex A) shall not be mandatory for a videotelephone terminal.

Consequently, tests will be made with a bit rate of 48 kbit/s if this rate is implemented or, oppositely, with a bit rate of 56 kbit/s.

All knowledge concerning tests and values for transmission characteristics is with regard to 64 kbit/s, and a certain amount of degradation is predictable on some characteristics (e.g. distortion and noise). Considering this the requirements shall correspond to the worst case (48 kbit/s).

4.1.2 Standard user's position

4.1.2.1 "Compact" terminal

In this type of terminal audio and video parts are inside the same housing.

The test position used in I-ETS 300 245-3 [4] does not appear to be realistic for a videotelephony terminal application.

Thus for normal use of videotelephony terminal, a standard user's position is considered, unless the manufacturer expressly recommends another position.

For simplification, all measurements shall be realised in conformance with I-ETS 300 245-3 [4] and a correction factor F shall be applied as defined and used below.

The value for this correction factor F shall be: $F(dB) = 20\log\frac{Ds}{0,5}$

Where D_S is the distance in meters between the terminal and user's position.

4.1.2.2 Terminal with separated acoustic front end

In this type of terminal, the acoustic front end with separated or joined microphone and loudspeaker is not in the same housing as the video part.

The supplier's declaration shall define the test position. Correction factor(s) F for sending and receiving shall be calculated on the basis of this declaration.

4.1.3 Volume control

Unless stated otherwise, the requirements shall apply for all positions of the user-controlled receiving volume control, if provided.

4.1.4 Test considerations

Unless stated otherwise, the conditions of testing for compliance shall apply according to annex A, clause A.1 of I-ETS 300 245-6 [1].

4.2 Sensitivity - frequency response

The requirements and test methods of I-ETS 300 245-6 [1] shall apply for the responses.

4.3 Sending and Receiving Loudness Ratings

Nominal values:

Sending Loudness Rating (SLR) = (12 - F) dB

Receiving Loudness Rating (RLR) = (6 - F) dB

The RLRmin, measured with the manual volume control at the maximum position, shall be:

RLRmin = (- 4 - F) dB

A manufacturing tolerance of ± 3 dB is allowed.

Compliance shall be checked by the test described in annex A, subclause A.2.2 of I-ETS 300 245-6 [1].

4.4 Volume control range

With a line level of - 15 dBm0 it shall be possible to obtain an RLR value which is at least 15 dB greater (quieter) than the RLR at - 30 dBm0 with manual and automatic gain control (if provided).

The acoustic output level shall be user controllable with a minimum range of 15 dB.

When a manual gain control is not used and if an automatic gain control is provided, the RLR value obtained with a line level of - 15 dBm0 shall not exceed that RLR value which is obtained with a line level of - 30 dBm0 by more than 15 dB. This avoids parts of negative law at the input/output characteristic.

4.5 Terminal Coupling Loss

4.5.1 Weighted Terminal Coupling Loss

The requirements and test methods from I-ETS 300 245-6 [1] shall apply.

4.5.2 Stability

The requirements and test methods of I-ETS 300 245-6 [1] shall apply.

4.6 Distortion

4.6.1 Sending distortion

The sending Signal to Distortion (S/D) ratio is the ratio of the signal power of the measurement tone to the distortion power at the digital output.

The sending S/D ratio shall be above the limits given in table 1. Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Table 1: Limits for signal to distortion ratio

Tone input level dB rel. ARL	300 Hz dB	1 kHz dB	6 kHz dB
10		21,5	
3		32,0	
- 3	26,0	32,0	29,0
- 11		32,0	
- 18		32,0	
- 40		12,0	

The test methods of I-ETS 300 245-6 [1] shall apply.

4.6.2 Receiving distortion

The receiving S/D ratio is the ratio of the signal power of the measurement tone to the distortion power at the digital output.

The receiving S/D ratio shall be above the limits given in table 2.

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Table 2: Limits for signal to distortion ratio

Tone input level dB rel. ARL	300 Hz dB	1 kHz dB	6 kHz dB
8		21,5	
0	26,0	27,5	29,0
- 7		32,0	
- 13		32,0	
- 23		32,0	
- 30		27,0	
- 40		17,0	
- 50		7,0	

The test methods of I-ETS 300 245-6 [1] shall apply.

4.7 Out of band signals

The requirements and test methods of I-ETS 300 245-6 [1] shall apply.

4.8 Noise

The test methods of I-ETS 300 245-6 [1] shall apply.

4.8.1 Sending

The A-weighted noise produced by the set in the sending path shall not exceed -62 dBm0(A).

This measurement shall be made during idle mode.

4.8.2 Receiving

This measurement shall be made during idle mode.

4.8.2.1 A-Weighted noise

With the volume control set to the maximum, the noise level will not exceed (- 46 + F) dBPa (A).

4.9 Delay

The requirements of I-ETS 300 302-3 [5] shall apply.

4.10 Lip synchronisation

There are no normative requirements for lip synchronisation.

> NOTE: A good synchronisation between the video signal and the audio signal will be ensured if the difference between the delays of the two paths is less than 25 ms. This value can be checked by taking into account the nominal overall delay (in both directions) introduced by video encoding and decoding tested according to ITU-T Recommendation H.261 (see annex A).

4.11 Switching characteristics

Requirements and test methods given in subclause 5.10 of I-ETS 300 245-3 [4] shall apply.

Annex A (informative): Bibliography

For the purposes of this I-ETS, the following informative references have been given.

- ITU-T Recommendation H.261 (1993): "Video codec for audiovisual services at p x 64 kbit/s".
- CCITT Recommendation G.722 (1988): "7 kHz audio coding within 64 kbit/s".
- ETS 300 264: "Integrated Services Digital Network (ISDN); Videotelephony teleservice, Service description".
- ITU-T Recommendation G.122 (1993): "Influence of national systems on stability and talker echo in international connections".
- ITU-T Recommendation P.51 (1993): "Artificial mouth".
- ETS 300 111: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Service description".
- ITU-T Recommendation P.64 (1993): "Determination of sensitivity / frequency characteristics of local telephone systems".

History

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